DIGITAL MICROPHONES FOR HIGH RESOLUTION AUDIO

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Microphones with digital output format have appeared on the market in the last few years. They integrate the functions of microphone, preamplifier, and analogue-to-digital converter in one device. Properly designed, the microphone dynamic range can thus be optimally adapted to the intended application. The need to adjust gain settings and trim levels is reduced to a minimum. Dynamic range issues inside and outside the microphone are discussed. Advantages of digital microphones complying with AES 42, with a wide dynamic range and 24-bit resolution are shown.

INTRODUCTION

One should first define the term “digital microphone” in the context of this article. A possible classification could comprehend:

- a transducer where the underlying acoustical-mechanical-eletrical transduction principle contains a quantization,
- a combination of separate transducers, each responsible for certain quantization steps,
- a microphone integrating an analog-to-digital converter (ADC).

The first category describes the “purely digital” transducer. The first microphone by Philipp Reis [1], a single contact transducer, represented such a transducer, albeit with very low quality due to the 1-bit resolution. This is the only purely digital transducer known to the author.

In the second category we find e.g. an optical microphone, where the position-dependant displacement of a diaphragm is traced with distinct light rays. The reflected rays excite separate sensors, whose outputs are combined into a single signal [2]. Another, electrostatic transducer experiment shows the diaphragm as part of the ADC, as component for the electrical / acoustical summation in the feedback loop of a ΣΔ-converter [3]. To obtain dynamic ranges comparable to the 120-130 dB of standard analogue microphones, these principles would need to be scaleable over 6 orders of magnitude, a feat hardly achievable due to the extreme mechanical precision involved.

Current microphone technology thus focuses on the third category: microphones with integrated ADC. Here, a purist could further differentiate between

- microphones with ADC output modules,
- microphones with ADC in closest proximity to the transducer,

where the first subcategory would describe a complete microphone, just with an added ADC module; the second subcategory represents transducers where the transducing element itself is closely integrated with the analogue-to-digital conversion process. In the context of high resolution audio it will be clear that the preferred transducer should be of the electrostatic (condenser) type, as this principle still yields the highest performance regarding parameters like linearity, dynamic range and frequency range.

1 HISTORICAL DEVELOPMENT

Possibly the first realization, in 1989, incorporating an ADC in the same housing with an electro-acoustical transducer is mentioned in [4]. The corresponding electret condenser microphone by Ariel company was intended for use with the now defunct NeXT computer, with the then available 16 bit transducers and a stated dynamic range of 92 dB. A 1995 prototype by Konrath [5] put an ADC circuit inside the housing of a commercial microphone. It featured a 7-pin XLR-connector and dedicated supply, delivering a multitude of supply voltages to the circuit. A later commercialised version by Beyerdynamic (MCD100) simplified this set-up with the adoption of phantom power, similar in principle to P48 defined in IEC61938 [6], but adapted to the lower voltage and higher current requirements of ADC components. It already featured a gain ranging ADC, to be discussed later, and limited remote control functions (pre-attenuation) but yielded sub-optimal noise figures, compared to standard analogue microphones. Another proprietary solution was presented by Milab [7].

Although the mentioned developments could not fully compete technically with state-of-the-art analogue microphones, they were helpful in starting discussions amongst manufacturers on the future of digitisation in microphones. It was found that, before presenting microphones with digital output to a wide public, all questions of power supply, interfacing, connector types, remote control etc. should be put into a public standard, to allow future products to interconnect between manufacturers. Accordingly, the German DKE 742.6 committee served as a starting basis, then handing over to an AES standardization committee to publish the AES 42-2001 standard [8,9], currently revised to the 2006 edition. Almost ten international microphone manufacturers were actively or passively involved, guaranteeing a common consensus. First microphones complying with the new standard were presented in 2001, as a full-feature large diaphragm microphone [10], later followed by a measurement microphone [11] and small diaphragm capsule systems [12,13].

In contrast to the professional audio approach, trying to provide highest possible audio quality, recently other solutions have been presented, driven by computer technology, i.e. mainly USB-powered microphones, with currently in comparison very limited specification ranges [14].

2 REASONS AND REQUIREMENTS FOR DIGITAL MICROPHONES

Analogue output condenser microphones are now, 90 years after their invention by E.C. Wente, certainly a mature technology. In a professional set-up, with appropriate cabling and limited outside interferences, a very high dynamic range of up to 130 dB-A can be transduced [15,16]. To reduce effects of cable length and interferences on the comparatively small microphone output signal, preamplifiers are often located in close proximity to the microphones. In any case, proper level matching of all analogue components is necessary to guarantee optimal signal transmission, allowing for sufficient head-room and foot-room in the process. On the other hand, digital technology provides potentially loss-less transmission, once the analogue-to-digital conversion has taken place. Accordingly, the interest for microphones with digital output arose when high quality ADC technology became available, allowing conversion only minimally affecting microphone specifications.

Some of the requirements on digital microphones [8] later realized in the AES 42 standard [9] were

- physical layer interface & protocol compatibility: AES3 protocol with overlaid phantom power, using 3-pin XLR connectors,
- control information from the microphone: via user bits in the AES3 data stream,
- control information to the microphone: via low frequent modulation of the phantom power voltage.

With the chosen interface, loss-less transmission can be performed over approximately 100 m also with high-quality "analogue" microphone cable, approximately 300 m with AES3 "digital" cable. This compares well with typical values for high-quality analogue set-ups.

An essential point in digital technology is proper synchronization of all audio streams to a reference clock. In a minimal set-up a receiver can synchronize to a single microphone, although this would be in contrast to typical studio procedures, where either the mixing console, or a dedicated reference clock provide the clocking reference. But, with multiple digital microphones one needs to either work with sample rate converters in every channel at the receiver side (AES42 mode1), or preferably synchronize the microphones to the reference clock (AES42 mode2). High quality sample rate converters do increase the cost, and even though in their current embodiments [17,18] they might not influence the signal much, they will increase processing time and thus add to the overall latency, which can become prohibitive in some applications, e.g. where direct monitoring is called for.

Sending the clock signal directly to the microphone would imply multi-lead cables, incompatible with standard 2-wire+ground/return studio wiring. The solution adopted by AES42, after extensive tests, was to integrate a voltage controlled crystal oscillator (VCXO) inside the microphone, yielding an already very stable data stream but where the frequency is dynamically fine tuned from the receiver side via the control information sent to the microphone (Fig. 1).

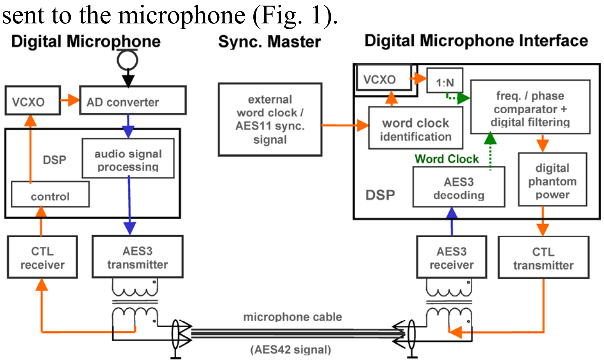

Figure 1: Connection of a digital microphone, with synchronization using AES42 interface specification. Microphone sample rate is controlled (CTL), comparing extracted microphone rate and external word clock.

The essential requirement then for digital microphones remains to integrate A-to-D conversion providing dynamic range and resolution comparable to their high quality analogue counterparts.

3 DYNAMIC RANGE AND NOISE

In order to be able to compare possible benefits of analogue and digital microphones, one has to look at the limiting factors, i.e. the behaviour at very small and large signal levels, corresponding to the noise floor and the overload characteristics, as well as the typical signal resolution, with a medium level signal present.

As mentioned, the typical dynamic range of the output of a condenser microphone capsule can exceed 130 dB, with typical maximum levels at a surprisingly high +10 dBu (2.5 V_{RMS}) and microphone self noise at -120 dBu (A-weighted). In the most noise free of current studio microphones this corresponds to sound pressure levels of 7 to 137 dB SPL, covering the needs of most applications. Only in excessively loud settings will there be a need to (manually) switch the pre-attenuation on, shifting the microphone's dynamic range to higher levels.

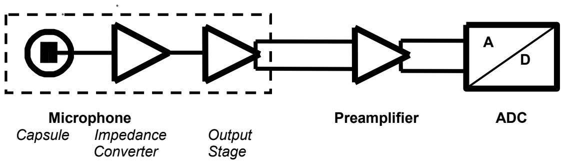

Figure 2: Simple analogue signal chain, with condenser microphone.

The typical noise voltage n_{mic} of a condenser microphone in Fig. 2 roughly follows a pink noise characteristic, whereas dynamic microphones, preamplifiers and AD converter inputs produce basically white noise. Preamplifier equivalent input noise n_{pre} (EIN) depends on the amount of gain v chosen. Concentrating all necessary gain inside the preamplifier, the sum of analogue equivalent input noise in an analogue recording chain with ADC will be

$$n_{sum,ana} = \sqrt{n_{mic}^2 + n(v)_{pre}^2 + n_{ADIn}^2 / v^2} \quad . \tag{1}$$

The physical limit for preamplifier noise is determined by the thermal noise of the input load R_i

$$n_{pre,\min} = \sqrt{4kTR_i\Delta f} \tag{2}$$

with k = 1,38*10-23 J/K (Boltzmann constant), T as temperature, and Δf as the bandwidth. For a typical microphone output impedance of R_i = 200 Ω, n_R calculates to -129 dBu (Δf = 23 kHz), or -131.7 dBu-A. At high gain settings many preamplifiers show noise figures close to this physical limit, but at low gain settings n_{pre} might be as high as -100…-80 dBu, and is seldom published in the specifications. One sees that preamplifier noise is higher or lower than the above mentioned microphone self noise of -120 dBu-A, and one main task for the recording engineer is then to optimise this sum, keeping preamplifier and ADC input headroom in mind. In analogue set-ups, the rule is to pull up the gain to studio reference level, trying to avoid clipping or distortion even with unforeseen very high sound pressure levels.

The working dynamic range of a typical microphone / preamplifier combination is shown in Fig. 3. The output level of the preamplifier $U_{out,pre}$ is shown over gain v. ADC noise is left out, for simplification, and assuming that the preamplifier gain will be optimally set, so that microphone and preamplifier noise dominate. The limitations are then given by:

- $n_{200\Omega}$: -131.7 dBu-A thermal resistive noise as physical limitation,
- n_{pre}: preamplifier equivalent input noise (A-weighted),
- Max_{pre}: maximum preamplifier output level, here: +20 dBu
- n_{mic}: microphone self noise, here: -120 dBu-A
- Max_{mic}: maximum microphone output level, here: +6 dBu

One sees that the preamplifier noise n_{pre} reduces the maximum dynamic range of the microphone *Dyn(Mic)* by approx. 16 dB, to a maximum resultant working dynamic range *Dyn(Max)* of 110 dB. At the upper/right axis the diagonal curves of constant equivalent input sound pressure level are given values, for a microphone with sensitivity M_0 = 12mV/Pa.

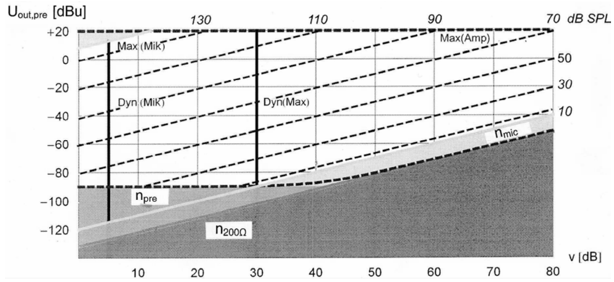

Figure 3: Dynamic range of a combination analogue microphone / preamplifier

The situation is different in the case of digital microphones with integrated ADC, as in Fig. 4. The capsule parameters can be chosen by the designer so that the capsule output levels are perfectly matched to the ADC input requirements. The noise sum then reduces to

$$n_{sum,dig} = \sqrt{n_{mic}^2 + n_{ADIn}^2} \quad . \tag{3}$$

Accordingly, the curve for preamplifier noise in Fig. 3 is replaced by the ADC noise n_{ADIn}. The noise over gain diagram for digital microphones is shown in Fig. 5. The dynamic range is vastly increased, especially for the small gain values often used with condenser microphones, and most importantly becomes independent of the chosen gain setting. It is now only limited by the microphone specifications, and by the digital processing limits, i.e. 0 dBFS level.

Note: The gain shown in Fig. 5 is performed after the ADC, i.e. in the digital domain.

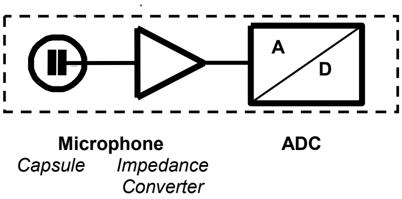

Figure 4: Condenser microphone, with integrated ADC

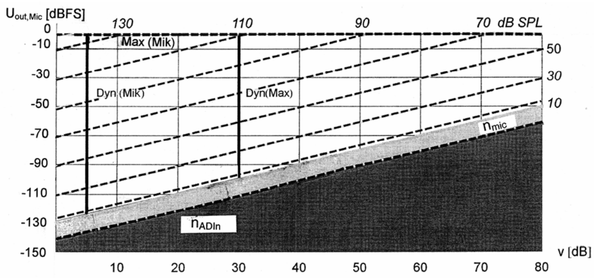

Figure 5: Dynamic range of a digital microphone

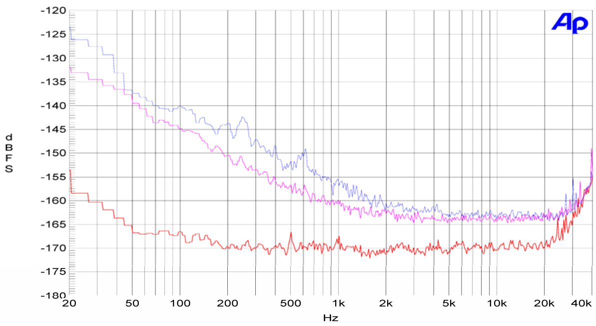

Figure 6: Noise spectra (16k samples, 32x averages) of an ADC with a. input short-circuited (-140 dBFS-A, lower curve), b. impedance converter and equivalent capsule capacitance (-133 dBFS-A, middle curve), c. impedance converter and real capsule (-130 dBFS-A, upper curve).

A more detailed perspective of the noise components is presented in the spectra of Fig. 6. With the input short-circuited, the ADC shows a roughly white noise characteristic n_{ADC}, typical of today's ΣΔ –ADCs, with slightly increasing noise above 20kHz, due to noise shaping algorithms. Reduced to a single value, the shown noise is in the region of -140 dBFS-A. The analogue impedance converter, loaded by a typical equivalent capsule capacitance, overlays this with a noise n_{ADIn} approx. 7 dB higher, yielding -133 dBFS-A. Adding a real condenser capsule, the thermal/acoustical capsule noise n_{caps} adds another 3 dB (-130 dB-A). This means that the thermal/acoustical noise n_{caps} of the capsule and the electrical noise n_{ADIn} of the combined impedance converter and ADC are roughly at the same level. To achieve even lower values, one would thus have to work on optimising both electronics and capsule.

As a side effect, the benign noise of the analogue components, capsule and impedance converter, with its largely gaussian distribution serves as an efficient dither on the ADC quantization noise [19]. With typical capsule parameters of small and large diameter cond-enser capsules, the summed noise $n_{sum,dig}$ can be at a level of -122 dBFS or -130 dBFS (A-weighted), respectively.

4 ADC CHARACTERISTICS

Fig. 6 shows an ADC with dynamic range of 140 dB-A. ADC circuits matching such a vast dynamic range would be of the gain ranging type, combining two or more ADCs working at different signal levels. This is one realization of a floating point converter, with exponents of 2^0 and 2^4 [20,21].

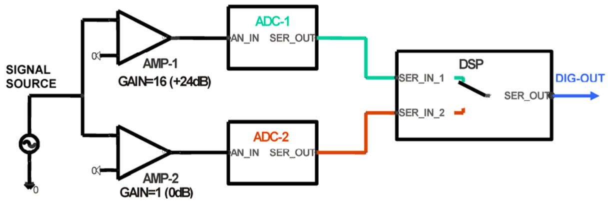

Figure 7: Simple gain ranging ADC circuit [10]

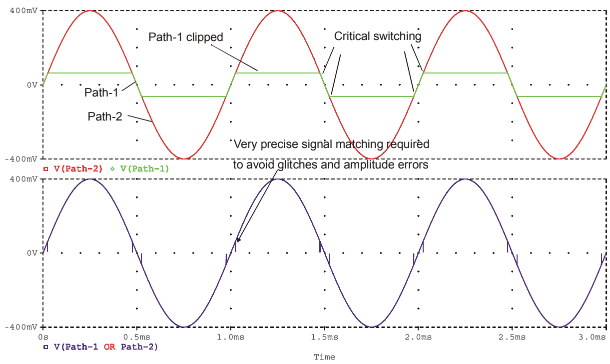

Figure 8: Signals in combined ADCs of Fig. 7, with audible "glitches" in the summed signal [10]

As is well known, switching directly between ADCs working at different levels can lead to artefacts like "glitches" (see Fig. 8), or noise modulation [20,21], when signal levels pass the switching level. The noise floor of an ADC is typically wide-band white noise. This white noise then becomes most audible when it is modulated by a low frequent signal, not masking the higher frequent white noise components. One possible way to reduce this effect is a non-linear network, keeping both ADCs always in operation, and summed, depending on the signal level, as shown in Fig. 9 & 10.

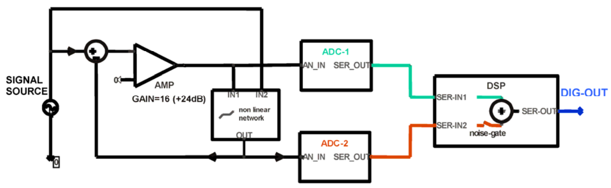

Figure 9: Gain ranging ADC circuit, with non-linear network [10]

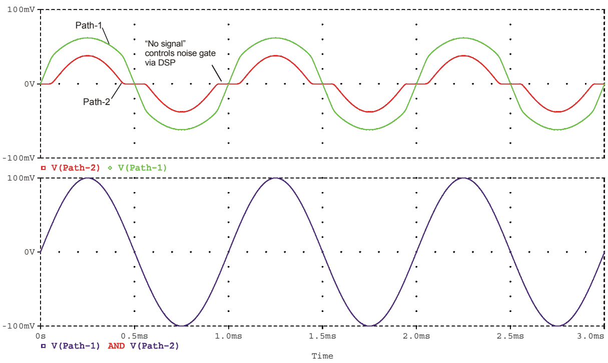

Figure 10: Separate signal paths and re-combination result in circuit of Fig. 9, with non-linear crossover topology [10]

As mentioned, the gain ranging ADC shown in Fig. 7 is a floating-point processor with exponents of 2^0 and 2^4. Combining them does widen the dynamic range by 4x6 dB = 24 dB, but does not improve their specific resolutions. Such a simple switching circuit will then modulate from the lower range ADCs noise to the higher range ADCs noise whenever the signal passes the crossover point, producing a distinct noise peak. A non-linear crossover network smooths this transition region out, making it inaudible. Properly designed, the result can then be a digital microphone with a dynamic range of up to 130 dB-A, with all noise components 80 dB below the signal over a wide dynamic range.

5 APPLICATION BENEFITS

From the above, some benefits for the user become immediately clear. With up to 130 dB-A, the dynamic range of the conversion covers the complete dynamic range of the analogue microphone counterpart. There is no need anymore for setting the gain controls in order to match input and output levels, as needs to be done with standard analogue recording set-ups. When recording to an appropriate 24 bit medium, the digital microphone can be connected and recorded directly, any gain levelling taking place after the recording, or just for monitoring purposes. The lower limit for the signals is determined by the self-noise of the capsule, thus by unavoidable physics, and the maximum allowed sound pressure levels cover the vast majority of applications. For very loud signals, the dynamic range of the capsule output and thus of the complete digital microphone can be shifted by e.g. 6, 12, or 18 dB with the same mechanisms as in analogue microphones (shunt capacitance, negative feedback, or reduced polarization voltage). For safety purposes, an additional very fast look-ahead peak limiter (see Fig. 11) implemented inside the microphone takes care of unforeseen excessive sound pressure levels.

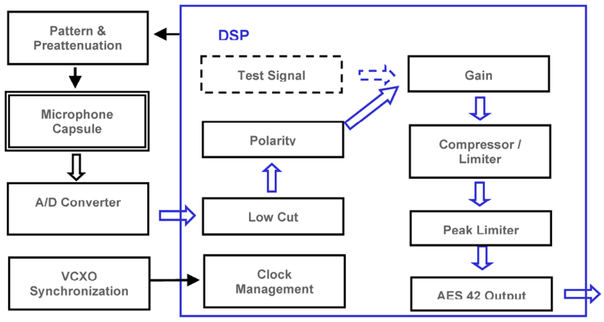

Figure 11: Signal flow in a digital microphone, with compressor and peak limiter

All this holds of course only true for the described professional digital microphones with very wide dynamic range, which the AES42 standardization committee had in mind. Other recent microphones with digital interface, powered by USB, show a very limited dynamic range, often with a noise floor consisting of undithered ADC quantization noise plus power supply artefacts, and thus offer no advantage over their analogue counterparts, other than simple connectivity to PC environments [14].

One side note has to be included, regarding current digital recording and monitoring equipment: Often, these devices are so designed as to expect only digital input signals aligned close to reference studio level, and accordingly only offer limited gain manipulation, e.g. +10dB, of such digital signals. As has been shown in Fig. 5, digital microphones can be recorded directly with the widest dynamic range if they are operated with no or small digital gain and do not require pulling up the gain as high as possible. Still, and be it only for direct monitoring purposes, those perfectly recorded low-level signals need to be made audible. It would be helpful then, to find more digital recording equipment offering amplification of digital input signals, and not only the analogue ones, over a wider gain range.

6 OUTLOOK AND CONCLUSION

Microphones with digital output are a comparatively new concept. Still, they show clear advantages regarding gain settings and dynamic range handling, and they are bound to find wide spread use. As the signal is transformed with a high-quality AD conversion to the digital domain, it is now also possible to obtain high-quality recordings with comparatively inexpensive, semi-professional recording equipment, if it does allow 24 bit word length, with the chosen sample rate. Digital microphones will make the job of the studio or location sound engineer simpler, reducing the probability of errors, thus keeping his mind free to concentrate on the acoustical and artistic aspects of the recording.

REFERENCES

[1] Reis JP (1861) Über Telephon durch den galvanischen Strom, *Jahresber. d. Physikal. Vereins zu Frankfurt am Main* (1860-1861) pp. 57-64

[2] Keating DA (1994) "Optical Microphones", in: Gayford ML (1994) *Microphone engineering handbook*, Focal Press, Oxford

[3] Yasuno Y, Riko Y,"A basic concept of direct converting digital microphone", *Acoust. Soc. Am. J.*, vol. 106, pp. 3335-3339 (1999)

[4] Paul JD et. al. (1991) "Digital output transducer", Patent US 5051799

[5] Konrath K (1995) "Konzeption und Entwicklung eines Prototyps des digitalisierten Mikrophons", Diplom Arbeit, FH Düsseldorf, Fachbereich Medien

[6] IEC 61938 "Audio, Video and Audiovisual Systems – Interconnections and Matching Values – Preferred Matching Values of Analogue Signals"

[7] Almeflo PO, Johansson M, "Suppression of switch mode power supply noise in digital microphones", preprint no. 5341, *110th AES Conv.,* Amsterdam (2001), see also www.milab.com

[8] Harris S et al., "Towards a digitally interfaced microphone standard", preprint no. 4518, *103rd AES Conv.,* New York (1997)

[9] AES42 standard for acoustics – "Digital interface for microphones"

[10] Peus S, Kern O, "The digitally interfaced microphone – the last step to a purely digital audio signal transmission and processing chain", presented at *110th AES Conv.*, Amsterdam (2001), downloadable at: www.neumann.com

[11] Microtech Gefell, Brochure: "MV230 digital - Digital measurement microphone", www.microtechgefell.com

[12] Schoeps (2006), Brochure: "CMD series", www.schoeps.de

[13] Robjohns H (2006) "Neumann KM184-D", *Line-Up* (Nov/Dec 2006); also: Brochure: "KM-D", www.neumann.com

[14] Funke R, "Untersuchungen an Mikrofonen mit digitaler Schnittstelle auf ihre Einsatztauglichkeit in Rundfunkproduktionen", Dipl.Arbeit, Fachhochschule Deggendorf, to be published (2007)

[15] Peus S, "Measurements on studio microphones", preprint no. 4617, *103rd AES Conv.*, New York (1997)

[16] Schneider M, "Eigenrauschen und Dynamikumfang von Mikrophon und Aufnahmekette", *20. Tonmeistertagung*, Karlsruhe (1998)

[17] Harris S et al., "A monolithic 24-bit, 96-kHz sample rate converter with AES3 receiver", preprint no. 4965, *106rd AES Conv.*, Munich (1999)

[18] McLaughlin KJ, Adams R, "An asynchronous sample rate converter with 120 dB THD+N supporting sample rates up to 192 kHz", preprint no. 5191, *109rd AES Conv.*, Los Angeles (2000)

[19] Vanderkooy J, Lipshitz SP, "Resolution below the least significant bit in digital systems with dither", *J. Audio Eng. Soc.*, vol. 32, pp. 106-113 (March 1984)

[20] Blesser BA, "Digitization of audio: a comprehensive examination of theory, implementation, and current practice", *J. Audio Eng. Soc.*, vol. 26, pp. 739-771 (October 1978)

[21] Fielder LD, "The audibility of modulation noise in floating-point conversion systems", *J. Audio Eng. Soc.*, vol. 33, pp. 770-781 (October 1985)